Some Design Issues in Local Multipoint Distribution Systems

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ABSTRACT: *Local multipoint distribution systems (LMDS) refer to millimeter-wave point-to-multipoint radio networks which were originally intended for digital TV broadcasting, video-on-demand (VoD), and similar consumer services with limited interactivity. It was later recognized that LMDS systems have a strong potential to supply broadband services to both homes and businesses, and the interest gradually shifted toward these applications. In this paper, we first give an overview of LMDS systems, and describe the technical specifications elaborated by the Digital Video Broadcasting (DVB) project and the Digital AudioVisual Council (DAVIC) which form the technical basis of most LMDS system developments to date. Next, we discuss frequency allocation, frequency planning, and LMDS system design to transport symmetric or asymmetric traffic. Then, we address a number of basic design issues which include antenna sectorization and frequency reuse patterns, discuss the use of higher-level modulations to increase cell capacity, and compare different multiple access techniques.*

INTRODUCTION

Originally driven by digital TV applications, there has been a tremendous effort over the past few years to develop standards for digital broadcasting over a variety of media including satellites, cable networks, and microwave and millimeter-wave radio systems. Standardization was first initiated in Europe with the establishment of the Digital Video Broadcasting (DVB) project conducted under the auspices of the European Broadcasting Union. The DVB project was in charge of elaborating the commercial requirements and technical specifications of different broadcast technologies. The technical specifications elaborated by the DVB project were next passed to the European Telecommunications Standards Institute (ETSI) for further procedures toward publication of the standards. The DVB specifications for satellites and cable networks were released in December 1994 [1], [2]. Then, in addition to digital TV broadcasting over the terrestrial VHF and UHF channels [3], attention was turned toward microwave multipoint distribution systems (MMDS) which operate at frequencies below 10 GHz, and to local multipoint distribution systems (LMDS) which typically operate at millimeter-wave frequencies above 20 GHz.

Another international body which was set up to elaborate technical specifications for broadcast as well as for interactive services over cable networks, satellites, and radio systems is the Digital AudioVisual Council (DAVIC) which groups major network operators, service providers, and consumer electronics, telecommunications and computer industries. Although the DVB project started earlier and released its cable and satellite specifications before DAVIC, the lead was clearly taken by DAVIC for MMDS and LMDS [4]. The DAVIC and DVB specifications differ to some extent, but they are identical in most aspects, and therefore, we will often refer to them as the DVB/DAVIC specifications, while pointing out their occasional differences wherever needed.

The original driver of the specification work carried out by both organizations was digital TV broadcasting and interactive services (video-on-demand, pay-per-view, home shopping, internet access,...) to residential customers, and digital satellite and cable TV broadcasting services have been deployed for several years. The introduction of digital broadcasting services by MMDS and LMDS have been much slower, particularly due to the competition from direct broadcast satellites. But in addition to the originally intended residential market, LMDS systems have been recognized to be attractive to supply broadband data and telephony services to small- and medium-size businesses. With respect to wired networks, radio systems clearly have the advantage of quick deployment, and this is very appealing for new operators which have emerged or are emerging in Europe, North America, and other regions of the world.

The purpose of the present paper is to give a general overview of LMDS systems, discuss their potential to offer broadband services to homes and businesses, and present a number of design issues related to their implementations. We also discuss frequency allocation as well as frequency planning and reuse, and indicate some potential technologies for possible future evolutions. First, in the next section, we briefly describe the DVB/DAVIC specifications for LMDS. This description covers the downstream channel (from central station to subscribers) as well as the physical layer and the medium-access control (MAC) protocol for the upstream channel (from subscribers to central station).

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Section 3 discusses the frequency allocation in Europe and in the US, as well as frequency planning and LMDS system design to transport symmetric or asymmetric traffic. Section 4 investigates several basic design issues including cell geometries, antenna sectorization, frequency reuse, the use of higher-level modulation schemes to increase cell capacity, and the potential of other multiple access techniques than the time-division multiple access (TDMA) used in DVB/DAVIC specifications. In Section 5, we give a summary of the discussions and our conclusions.

DVB/DAVIC SPECIFICATIONS

Downstream Channel

The DVB specifications for modulation, channel coding, and related functions for the LMDS downstream channel [5] can be summarized as follows: The system uses a quaternary phase-shift keying (QPSK) modulation and a concatenated forward error correction (FEC) coding scheme with a convolutional inner code and a Reed-Solomon (RS) outer code. The transmission frame is based on the MPEG2 transport data stream [6], and prior to channel coding, a scramb- ler is used to randomize the input signal.

The outer code has a block length of 204 bytes, carries 188 information bytes, and can correct up to 8 byte errors per block. This code is obtained by shortening the RS(255, 239) Reed-Solomon code to a block length of 204. The associated bandwidth expansion is approx. 8.5 %. A convolutional interleaver [7] with interleaving depth of I = 12 is inserted between the inner and outer encoders in order to uniformly distribute the errors which occur by bursts at the Viterbi decoder output in the receiver. With this interleaving scheme, a 12-byte error burst at the Viterbi decoder output appears as 12 isolated byte errors with a spacing of 204 bytes at the RS decoder input, and the RS decoder corrects all of these errors. In fact, since the RS code employed can correct 8 byte errors per block, this interleaving scheme can handle error bursts of up to $8 \times 12 = 96$ bytes or 384 QPSK symbols. The inner code is a rate-1/2 convolutional code with constraint length K = 7 (the NASA code which has become a *de facto* industry standard [8]), but the DVB specifications also include higher code rates (2/3, 3/4, 5/6, and 7/8) by puncturing this basic code. This allows to trade off coding gain against useful data rate on a given link.

The same specifications were also adopted by DAVIC except that DAVIC allows a roll-off factor of 0.2 or 0.35 in the square-root raised-cosine transmit and receive filters. Further, DAVIC specifications also allow the use of the 16-state quadrature amplitude modulation (16-QAM) in addition to QPSK. More specifically, DAVIC defines two grades: Grade A uses QPSK only, whereas Grade B specifications include both QPSK and 16-QAM. Strictly speaking, the DVB specifications therefore appear as a subset of DAVIC specification, but the difference between the two is only minor, and the 0.2 roll-off and 16-QAM options can be viewed as future extensions of the DVB specifications.

DAVIC specifications also include a mapping function to map ATM data on the MPEG2 transport stream originally defined for digital TV broadcasting. This function, which is illustrated in Fig. 1, is as follows: The incoming ATM data stream is partitioned into groups of 7 ATM cells, and each group is appended with 3 control bytes to form two consecutive 187-byte packets. Next, one sync byte is appended to each of these packets to form a 188-byte frame. Finally, 16 redundancy bytes are added to each frame for RS encoding, and this results in two consecutive MPEG2 transport stream frames.

Control byte	ATM cell 1 53 bytes	ATM cell 2 53 bytes	ATM cell 3 53 bytes	ATM cell 4 (part) 27 bytes

a) First packet

Control byte	Control byte	cell 4 (cont'd) 26 bytes	cell 5 53 bytes	cell 6 53 bytes	cell 7 53 bytes

b) Second packet

Fig. 1 : Mapping of 7 ATM cells onto 2 consecutive MPEG2 transport frames

Upstream Channel

The lead for the definition of a return channel was clearly taken by DAVIC not only for LMDS, but also for MMDS and cable networks. The reason for this is, as mentioned earlier, the DVB project was exclusively concerned with broadcast services in its first phase. Accordingly, we will focus here on the DAVIC upstream (return) channel specifications, which were also adopted with little or no changes by the DVB project.

As mentioned earlier, the multiple access technique used on the LMDS return channel is TDMA. The MAC protocol allocates time slots to different users, and each customer premises equipment (CPE) transmitter can transmit only when a time slot is allocated to it. The time slots are composed of 68 bytes which include a 4-byte preamble and a 1-byte guard interval at the end. The remaining 63 bytes include 53 information bytes, i.e., one ATM cell, and 10 parity-check bytes for the RS code employed. That is, the return channel employs an RS(63,53) code with 8-bit code symbols and a 5-byte error correction capability per block. Before RS encoding, the data packets appended by the preamble and the guard interval are randomized through a byte randomizer. The modulation scheme is a differentially-encoded burst-type QPSK. Channel filtering is of raised-cosine Nyquist type evenly split between transmitter and receiver. The roll-off factor is 0.3.

Clearly, error protection on the upstream channel is not as efficient as the concatenated coding scheme used on the downstream channel. In addition to coding itself, the bursty nature of traffic leads to further performance degradation. Indeed, the burst QPSK receiver at the central station will typically have a degradation on the order of 2 dB with respect to the continuous-stream

QPSK receiver of the user terminal. These fundamental differences can be compensated, however, by the design of transmit and receive functions on the upstream and downstream channels.

MAC Protocol

We will briefly discuss here the MAC protocol used to allocate resources to different user terminals by the central station. Note that both the downstream and the upstream frames are divided into time slots that encapsulate exactly one ATM cell. Each frame on the downstream channel includes a frame start slot followed by random access slots which carry MAC messages and higher layer data. The upstream frame is divided into polling response slots, contention slots, and reserved time slots. The polling time slots are allocated to one subscriber terminal and may be utilized for a poll response after receiving a poll request from the central station. The contention slots are the time slots that are typically allocated to more than one terminal and utilization of contention time slots may cause a collision with another terminal trying to use the same slot. When a collision occurs, the contention may be resolved by a number of algorithms such as random retransmission delays which indicates to each terminal how many frames it has to wait before retransmission. Reserved time slots are reserved for use by only one terminal. The terminal transmits on these time slots whenever it has data to transmit. If no data is available, it transmits an idle cell. The contention and polling time slots are determined by the central station which decides which carrier frequency and time slots are to be used by each terminal.

The central station periodically polls each user terminal to establish, maintain, and terminate connections. The polls are periodically repeated at an interval of less than or equal to 2 seconds. It declares that the terminal is not responding if it receives no response to a polling request for 10 seconds. When a terminal attempts to enter the network, it acquires a downstream channel and listens for the poll directed to it. If it receives no poll request for 2 seconds, it switches to the next downstream channel and listens again. This process is repeated until the terminal finds the downstream channel on which it is being polled. The first task is to calibrate the user terminal in terms of clock phase so that it can transmit on poll response time slots without interfering with adjacent time slots and other terminals. In addition to the clock phase, the terminal also performs power control and carrier frequency control. Power control compensates for unequal signal attenuations resulting from different physical distances of user terminals to the central station on one hand and different propagation conditions on the other hand. This control loop sets and periodically updates the signal level transmitted by the terminal such that the central station receives a predetermined nominal signal level. Similarly, the user terminal performs carrier frequency control in order to compensate for the large frequency uncertainty and drifts of the microwave oscillator used in the transceiver which may be far beyond the capability of the demodulator.

LMDS NETWORKS

The first frequency allocations for LMDS systems were the 27.5 - 28.35 GHz band in the US and the 40.5 - 42.5 GHz band in Europe. Furthermore, these bands were initially intended for residential services such as TV broadcasting, video-on-demand (VoD), and similar consumer entertainment services. In fact, the 27.5 - 28.35 GHz band in the US was exclusively intended for downstream transmission, and therefore the 31.075 - 31.225 GHz band was next allocated to the upstream channel. This kind of asymmetry between the upstream and downstream directions is also present in the 40.5 - 42.5 GHz European band, since only a small portion of this band is allocated to the upstream direction.

Despite the availability of technologies and dedicated frequency bands, it is today questionable that massive deployment of LMDS systems can take place in the near future for digital TV broadcast and consumer entertainment services. The reason is that direct broadcast satellites are already in place, and it is quite unlikely that LMDS will be able to economically compete with satellite systems for these applications for the several years to come. Their first field deployments are therefore driven by broadband radio access services to small- and medium-size businesses, as well as to residential customers. These services include telephony, ISDN, high-speed internet access, and leased lines offering a variety of data rates. LMDS systems are also of particular interest to cellular and personal communication systems (PCS) operators for the interconnection of their base stations to the fixed public switched telephone network (PSTN) and to existing data networks.

Frequency allocation for LMDS is currently evolving so that broadband telephony and data services can be offered to both business and residential users. For example, the 24.5 - 26.5 GHz and 27.5 - 29.5 GHz ETSI bands are now open for these systems in addition to their conventional use for point-to-point systems. Also, the use of the 28 GHz band in the US has also changed in the sense it can be used not only to offer broadcast services, but any kind of symmetric or asymmetric services.

In Europe, LMDS systems will use the same frequency channeling as conventional point-to-point radios. Specifically, operators can use any channel spacing obtained through successive divisions by 2 of the basic 112 MHz spacing. That is, any of the following channel spacings can be used: 112, 56, 28, 14, 7, and 3.5 MHz. Furthermore, each LMDS cell is typically divided into a number of sectors each served by a separate central station antenna.

Suppose that each cell is divided into four 90° sectors and that in the downstream direction, a 28 MHz channel is allocated to each sector. With a roll-off factor of 0.35, the maximum symbol frequency without adjacent channel interference is 20.74 Mbaud. With the QPSK signal format, this gives a total bit rate of 41.48 Mbit/s on a 28 MHz channel. Now, recall that in the DVB/DAVIC specifications each 204-byte frame includes 16 redundancy bytes for RS encoding and 1 sync byte, and therefore the payload is only comprised of 187 bytes. Furthermore, mapping of input ATM cells is such that 7 consecutive ATM cells are mapped onto two consecutive MPEG2 transport frames leaving 3 control bytes. The total redundancy in these operations is therefore (2x17+3)/408 = 9 %. The net bit rate in this example is therefore 37.72 Mbit/s. This indicates that a 2x16 Mbit/s signal can be transmitted on a 28 MHz downstream channel without any problems. In the sequel, we will assume that a 28 MHz is allocated and a 2x16 Mbit/s net bit rate is transmitted in the downstream direction of each sector.

The channel spacing is significantly smaller on the upstream channel in order to reduce the cost of subscriber terminals. Suppose that the upstream channel spacing is 7 MHz. Each one of those channels can transmit a data rate of 4x2 Mbit/s. Obviously, 4 upstream channels need to be assigned to each sector in the example at hand to have an upstream capacity on the same order of magnitude as the downstream capacity of each sector. Under this assumption, each cell has a transmission capacity of 64x2 Mbit/s in each of the upstream and downstream directions and uses a frequency bandwidth of 224 MHz. Further, as it will be shown in the next section, coverage of a given geographical area can be made with a frequency reuse factor of 1 between cells, and therefore, a 224 MHz bandwidth is all that is needed to cover a whole region if a single downstream channel is assigned to each sector.

The next thing to point out is that the cell radius in LMDS systems typically does not exceed 2 or 3 km. Assuming a 2.5 km radius, the above example gives a capacity of 6.5 Mbit/s per km^2. If a higher capacity is needed, then more than one downstream carrier needs to be assigned to each sector. The capacity becomes 13 Mbit/s per km^2 with 2 channels per sector, and 19.5 Mbit/s per km^2 with 3 channels per sector, and so forth.

SOME DESIGN ISSUES

Frequency Reuse

We will now discuss the possible frequency reuse plans in a given geographic area. A first possibility is to use rectangular cells with 90° sector antennas at the central station [9] as shown in Fig. 2. Each quadrant of a cell in this figure is labelled with a digit which indicates the frequency (or group of frequencies) used in that sector. Note that all cells use the same frequencies, which indicates that covering of a region only requires 4 times the frequency bandwidth used in one sector. Subscriber terminal antennas are highly directional and point toward the central station serving their sector. Antenna sectorization within a cell rather than splitting each cell into further cells to increase capacity has the advantage of reducing maintenance cost in addition to easing network upgrade [10]. The reason is that all equipments serving a cell are located in the same place with antenna sectorization, whereas cell splitting requires to install equipments in the centers of the newly defined smaller size cells and to make further connections to the fixed network.

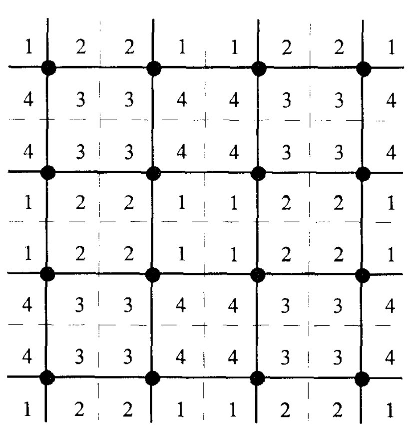

Fig 2 : Rectangular cell pattern with 90° sectors

Assuming that all central station transmitters have the same transmit power and that there is a perfect transmit power control at user terminals, the worst-case carrier-to-interference (C/I) ratio in these rectangular frequency reuse patterns is easily shown to be C/I = 20log(5) = 14 dB. First, focusing on downstream transmission, a user located on a diagonal passing through a set of central stations receives a useful signal from the central station serving its sector and interference from central stations that use the same frequency allocation in the same directions. These central stations are located at distances of 4D, 8D, 12D,... from the central station at hand, where D designates the half-distance between two adjacent central stations located on a diagonal. Now, if the user terminal is located at a distance d from the serving central station, the C/I ratio is given by C/I = 20log{(4D+d)/d}, where we neglected interference from central stations located at a distance of 8D or higher. It is easily seen that this expression is minimized and gives C/I = 14 dB for d = D. It is also easily verified that the same expression equally holds for upstream transmission.

Another possibility is to use hexagonal frequency patterns. Three types of sectorizations, particularly suitable for hexagonal cells, are illustrated in Figs. 3-5.

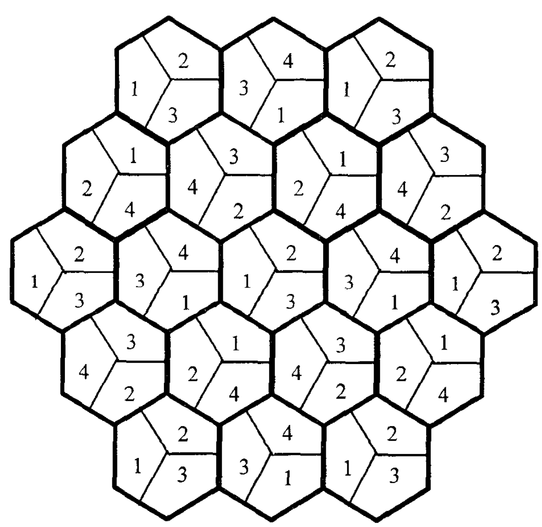

Fig. 3 : Hexagonal cell pattern with 120° sectors

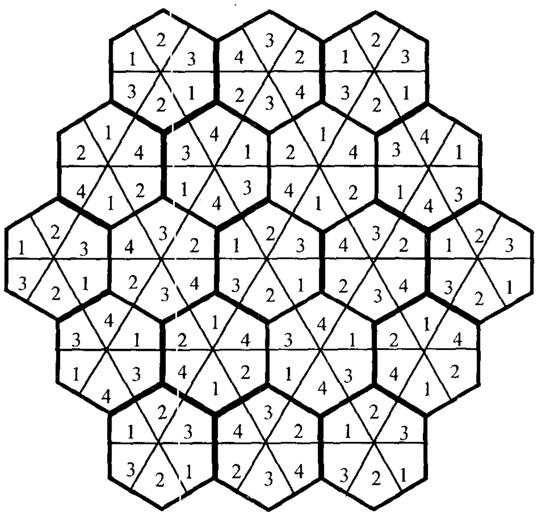

Fig. 4 : Hexagonal cell pattern with 60° sectors

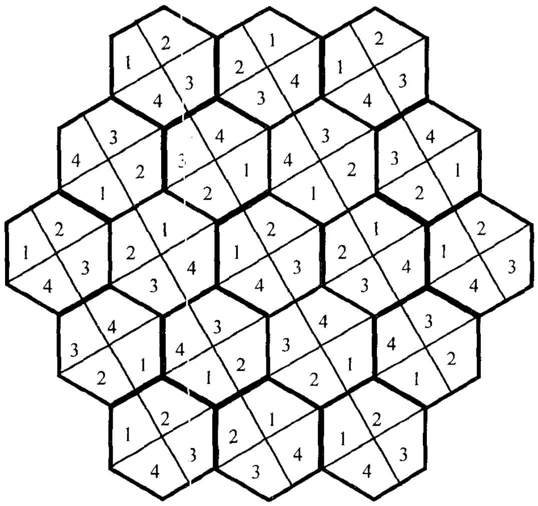

Fig. 5 : Hexagonal cell pattern with 90° sectors

In these figures, each sector is labelled with a digit which indicates the channel allocated to it. As can be seen, Fig. 3 employs 120° sectors, Fig.4 employs 60° sectors, and finally Fig. 5 employs 90° sectors. In all of these frequency reuse patterns, the reuse factor is 4 between sectors, and the minimum distance between central stations which have the same frequency allocation in the same directions is 4D, where D designates the cell radius. As in the rectangular frequency pattern of Fig. 2, the worst-case interference occurs for users located at a distance D from the serving base station and such that the serving station and the closest interfering station are exactly in the same direction from the user. Given the distance of 4D between closest interfering cells, the worst-case C/I ratio is 14 dB in all of those hexagonal patterns, which is identical to that of rectangular cell patterns of Fig. 2.

The advantage of hexagonal cell patterns over rectangular cells is a better use of the avilable transmit power in a given area. Indeed, a simple inspection shows that two circles centered on two adjacent central stations and fully covering their respective cells have a larger overlap in rectangular cell design. This implies that for equal distance between adjacent central stations and equal performance for users located at the furthest points of the cells from their central station, a lower transmit power is needed in hexagonal cells.

Higher-Level Modulations

First generation LMDS systems will be based on QPSK which does not make an efficient use of the available spectrum compared to the QAM signal formats with a higher number of states. This choice is perfectly justified by the current state of technology and low-cost objectives, but there is no doubt that the trend of ever-increasing capacity in communication networks will continue in the future, and this will naturally lead to the use of higher-level modulations.

The cell capacity can be doubled by using 16-QAM and tripled by going to 64-QAM. These modulations are suited for the downstream channel which transports a continuous data stream. But with TDMA as multiple access technique on the return channel, upstream modulators and demodulators operate in burst mode, and coherent detection requires a significant amount of overhead which reduces the system capacity. Therefore, rather than the conventional QAM signal constellations whose states are on a square grid, differentially-encoded amplitude and phase shift keying (DAPSK) signal constellations [11] which easily lend themselves to noncoherent detection are appealing for this application.

The 16-state DAPSK signal constellation is sketched in Fig. 6. It consists of two sets 8 signal point located

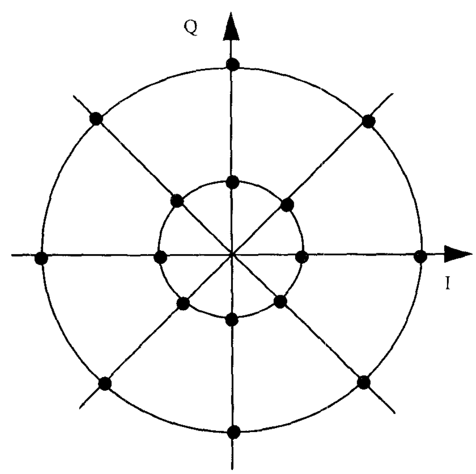

Fig. 7 : 16-DAPSK signal constellation

on two concentric circles. Each of those sets forms an 8-state phase shift keying (8-PSK) signal constellation, and the two sets are perfectly phase aligned. Since the signal constellation has 16 states, it carries 4 bits per symbol. From those, 3 bits are carried by phase transitions (as in differential 8-PSK), and 1 bit is carried by amplitude transitions. Provided that $\alpha > 1+2\text{Sin}(\pi/8)$, the average power of this constellation is related to the minimum distance through the relation for 8-PSK. From

$$P = \frac{1+\alpha^2}{8Sin^2(\pi/8)} d_{\min}^2 . \quad (1)$$

This is to be contrasted to $P = 5d_{min}^2/2$ for 16-QAM, and to

$$P = \frac{1}{4Sin^2(\pi/16)} d_{\min}^2 \quad (2)$$

these expressions, it is easily seen that in terms of minimum distance, 16-DAPSK loses 1.5 dB with respect to 16-QAM and gains 2.7 dB with respect to 16-PSK. In terms of sensitivity to phase noise, it is similar to 8-PSK, i.e., it is more robust than both 16-QAM and 16-PSK.

When both directions of the transmission are upgraded to the same modulation levels (for example the downstream channel to 16-QAM and the upstream channel to 16-DAPSK), the system remains symmetric (assuming the redundancy is the same in both directions). If only the downstream channels are upgraded, or if the upgrades are not carried out symmetrically, the downstream will have a higher capacity than the upstream. For example, if downstream channels are upgraded to 64-QAM and upstream channels are unchanged, the downstream direction will have 3 times the capacity of the upstream direction. One way to make the system symmetric in this situation is to change the frequency allocation plans so as to assign a higher bandwidth to the upstream channel. More precisely, three quarters of the total available bandwidth need to be allocated to the upstream direction to make the system symmetric in the case at hand. The overall system capacity increase is then 50 %. Next, suppose that downstream channels are upgraded to 64-QAM and upstream channels are upgraded to 16-DAPSK. In that case, 40 % of the available bandwidth needs to be allocated to downstream channels, and 60 % of it to the upstream channels in order to keep the same capacity in both directions. The network capacity in this scenario is increased by 140 %.

Multiple Access Techniques

As mentioned previously, the multiple access technique used in current LMDS systems is TDMA. More precisely, TDMA is used to share resources of each upstream carrier, but each user has access to the resources of all carriers allocated to its sector. In other words, the multiple access scheme is a combined TDMA/FDMA. An obvious question is whether other multiple access schemes such as code-division multiple access (CDMA) or orthogonal frequency-division multiple access (OFDMA) [12] offer new perspectives to the development of those systems in the future.

First, let us analyze the potential of CDMA. This technique, which originates from spread-spectrum technology, is used in the North American cellular radio standard IS-95 [10], and has recently been adopted for other future cellular and satellite systems. Comparison of TDMA and CDMA has been a very controversial subject often dominated by commercial interests, and therefore, it is difficult to find a truly objective comparison in the literature. Another difficulty is the fact that comparisons are often between systems in which TDMA or CDMA is only one ingredient among many others.

There are basically two classes of CDMA: Orthogonal CDMA (OCDMA) which employs a set of orthogonal sequences, e.g., Walsh-Hadamard sequences, and nonorthogonal CDMA which employs pseudo-noise sequences. Practical systems often use a combination of these two techniques. For example, in the IS-95 mobile radio standard, OCDMA is used to share resources in each cell, and on top of these ortogonal spreading sequences, long pseudo-noise sequences are overlayed to separate signals of different cells. First, it is not difficult to demonstrate that OCDMA has identical capacity to TDMA. If W designates the bandwidth required by one user, N users can be accommodated in both TDMA and OCDMA when the total available bandwidth is N.W. TDMA accommodates these users by allocating different time slots, and OCDMA acommodates them by allocating mutually orthogonal spreading sequences. Since the number of orthogonal sequences of length N is exactly N, OCDMA accommodates exactly the same number of users as TDMA. The other class of CDMA, i.e., pseudo-noise CDMA (PN-CDMA) is more difficult to evaluate, because the capacity is not a fixed number in this case. In this class of CDMA, all users interfere with each other, and capacity depends on how much interference (and performance degradation) one is prepared to tolerate. Note that in PN-CDMA a given user gets an interference level of 1/N (if the useful signal level is normalized by 1) from each other user of the same cell. With N active users in the cell, $C/I = N/(N-1) \cong 1$. If the interference level is to be kept below 25% of the useful signal level, then only N/4 users can be accommodated. This implies that even with a reuse factor of 1 between cells, PN-CDMA achieves a lower-capacity than OCDMA and TDMA employing a frequency reuse factor of 4 if the interference level is to be kept below that value. In this example, the C/I ratio of 4 in PN-CDMA only accounts for intracell interference. In other words, the C/I ratio will be much worse in practice due to intercell interference. For this reason, practical CDMA systems typically employ OCDMA within cells in order to suppress intracell interference, and PN-CDMA between different cells, but even then it is questionable whether CDMA can achive the capacity of TDMA.

Despite these negative capacity arguments, CDMA has a significant advantage over TDMA in terms of peak-to-average signal power, and this makes it appealing for use on the upstream channels. For a given average transmit power of the CPE, the peak power is N times larger in TDMA, because this multiple access technique concentrates the transmitted signal energy on the allocated slots, whereas CDMA spreads it over the entire frame. This holds when TDMA employs only one slot per frame and CDMA allocates one sequence only. If the user requires higher resources, TDMA will allocate a higher number of time slots per frame, and CDMA will allocate several spreading sequences. In that case, the advantage of CDMA over TDMA will be diminished, but still CDMA has a significant advantage which can be exploited to make low-cost user terminals. The situation is exactly the opposite for the central station, because if the transmitted signal is QPSK in time-division multiplexing (TDM), it is a sum of QPSK signals with a higher peak-to-average power in code-division multiplexing (CDM).

As for OFDMA, this multiple access technique was proposed for use on narrowband interference channel such as the return channel of CATV networks. OFDMA yields the same capacity as TDMA and OCDMA on Gaussian noise channels, but can support much higher levels of interference than TDMA and CDMA. Further, the reduced peak-to-avearage power ratio of CDMA also applies to OFDMA, which makes it attractive for user terminals. Finally, channel equalization is much easier in the case of OFDMA, which is reduced to multiplying by a complex coefficient each demodulated carrier at the receiver. No equalization is needed at all if the modulation is QPSK and detection is made differentially. The disadvantage of OFDMA is its higher sensitivity to phase noise which makes it necessary to use highly stable low-noise oscillators in the modulator.

Based on this analysis, it may be concluded that for the downstream channel there is no better multiplexing technique than the TDM adopted in the DVB/DAVIC specifications, because neither CDM nor orthogonal frequency-division multiplexing (OFDM) have any potential advantages in terms of capacity, and both of them suffer from increased peak output power. For the upstream channel, however, both CDMA and OFDMA lead to reduced transmit peak power with respect to TDMA, a property which eases the design of low-cost user terminals.

CONCLUSIONS

We first described the DVB/DAVIC specifications which form the technical basis of most LMDS developments to date, and pointed out that the scope of LMDS has significantly changed over the past few years shifting from the originally intended digital TV broadcast and related interactive consumer entertainment services to broadband data services for both business and residential customers. We highlighted the potential of LMDS systems for new operators due to their ease of deployment and the independence that they insure from the long-established national or regional operators which own a wired access network.

Next, we discussed a number of design issues related to the deployment of first generation LMDS systems and their future evolutions. This includes frequency reuse in a given geographic area, the use of higher-level modulations to increase cell capacity, and multiple access using CDMA or OFDMA. After briefly describing the popular rectangular cell design with 90° sector antennas, we presented several hexagonal cell designs with different types of antenna sectorizations and a frequency reuse factor of 4 between sectors. Regarding higher-level modulations, we pointed out the potential of DAPSK signal constellations which easily lend themselves to differential detection. Finally, our analysis of different multiple access techniques led to the conclusion that CDMA has little, if any, to offer with respect to TDMA in terms of cell capacity, but that both CDMA and OFDMA may lead to lower-cost user terminals since they reduce the transmitted peak signal power with respect to TDMA.

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